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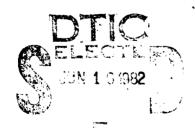


THE RECOGNITION OF WORDS FROM PHONEMES IN CONTINUOUS SPEECH

THESIS

AFIT/GE/EE/81D-9

Claude A. Baker Captain USAF



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THE RECOGNITION OF WORDS FROM PHONEMES IN CONTINUOUS SPEECH

THESIS

Presented to the Faculty of the School of Engineering
of the Air Force Institute of Technology
Air University

in Partial Fulfillment of the Requirements for the Degree of

Master of Science

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Captain

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Graduate Electrical Engineering

December 1981

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The work performed in this thesis is a first attempt at word recogniton based upon the phonemic output of the Air Force Institute of Technology (AFIT) acoustic analyzer. The acoustic analyzer is the result of Dr. Matthew Kabrisy's long term guidance, hopes, and energy and, oh yes, let us not forget the sleepless nights of many of his students.

The final report for several speech recognition projects which I encountered during my research contained phrases which, when paraphrased, become, "Given more time and money, the recognizer would have worked perfectly." Let me not be different. For many reasons, but primarily those of time and a lack of data to process, the vocabulary is limited to the 10 digits. The system has had insufficient training and experience to determine whether or not it will really work. Preliminary results are encouraging. The system has parsed all words thus far presented, but there have been, at most, only four utterances of each word incorporated into the state table.

My thanks to Dr. Kabrisky, Capain Larry Kizer, and Major Walt Seward. It has been an interesting though frustrating experience.

My thanks and love to my wife, Polly, who was always supportive and always acquiesed to my plaintive plea of, "I just need one more thing for my computer to finish the thesis." Also, my love to Yvette and Dane who patiently endured Daddy's studies and waited for tomorrows that rarely came.

Claude Baker

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Abstract

Thesis examines continuous speech recognition for a vocabulary of the digits 0 through 9. Recognition is accomplished by parsing phoneme strings obtained from the AFIT (Air Force Institute of Technology) acoustic analyzer. Word recognition is accomplished by a finite state automata implemented in UCSD Pascal. Novel features of the program include the ability to load and exercise state tables representing different vocabularies, as well as the ability to interactively edit and modify the state tables without recompiling. Complete details of the program are included.

The quality of recognition is indeterminate at this time. Sufficient training of the system has not been accomplished to determine validity of the recognition strategy. The system will parse all utterances for which it has been trained, although no more than four utterances of each word have been incorporated in the state tables.

THE RECOGNITION OF WORDS FROM PHONEMES IN CONTINUOUS SPEECH

I. Introduction

This thesis treats the topic of continuous speech recognition by machine. The ability to converse with machines through natural speech is a long sought goal of We dream of intelligent, omniscient companions; god-like in their power but existing merely to serve. literature of science-fiction is replete with examples of these machines. HAL, of the movie, "2001, A Space Odyssey", is one of the better known sentient computers that listens and talks. Though we desire these servants, we fear them. A recurring theme of the stories is, "Who will be master, man or machine?" Few of the super-computers of imagination have been benign. Many turned on their masters destroyed them, as did HAL. We love them and fear them, yet as scientists and engineers we continue to pursue the "intelligent" machines for the great benefits obtained.

Current news magazines and the trade magazines of the electronic industry seem to have frequent predictions or announcements of soon to be released equipment which will replace the office secretary. If we are to believe these

announcements, in the not distant future we will need no secretaries, nor grammar teachers, nor a wide range of other people who deal with words.

Background

The Information Processing Technology Office of the Advanced Research Projects Agency of the Department of Defense (ARPA) in November, 1971 initiated a speech understanding research (SUR) project to achieve computer understanding of human speech. The ARPA SUR project had a large budget, involved many separate groups, and spanned five years. Its goal was a major breakthrough in speech understanding capabilities (Ref 1:249-251). Now, ten years later, we have yet to attain meaningful, natural speech recognition by computer. The phrase natural speech recognition is used in the sense of continuous or conversational speech recognition.

The fact that the SUR project was sponsored by the Department of Defense is significant. The military has much to gain from achieving speech recognition capability. A wide range of tasks will benefit from the automation of speech recognition. Table I lists a set of tasks in which speech recognition would be useful to the military. More than any other task this thesis is directed toward system control through speech recognition.

Table I

- Speech Recognition Tasks with Military Application
- 1) Security
 - 1.1 Speaker verification
 - 1.2 Speaker identification
 - 1.3 Determining emotional state of speaker
 - 1.4 Recognition of spoken codes
 - 1.5 Secure access voice identification
 - 1.6 Surveillance of communication channels
- 2) Command and Control
 - 2.1 System control (ships, aircraft, fire control, etc.)
 - 2.2 Voice-operated computer interfaces
 - 2.3 Data handling and record control
 - 2.4 Material handling (mail, baggage, publications, industrial applications)
 - 2.5 Remote control (dangerous material)
 - 2.6 Administrative record control
- 3) Data Transmission and Communication
 - 3.1 Speech synthesis
 - 3.2 Vocoder systems
 - 3.3 Transmission data rate reduction
 - 3.4 Ciphering, coding, scrambling
- 4) Processing Distorted Speech
 - 4.1 Diver (helium) speech
 - 4.2 Astronaut communication
 - 4.3 Underwater telephone
 - 4.4 High "G" force speech

(Ref 1:310)

Problem

The research objective of this thesis was to develop an algorithm to recognize words from phoneme strings in continuous speech. The phoneme strings were to be provided

by an acoustic processor which has been under development at the Air Force Institute of Technology (AFIT) for several years (Ref 2).

Felkey (Ref 2) indicated that the acoustic processor was 70% to 90% accurate in locating and identifying phonemes and presented results to support the figures. Based upon these figures, an existing algorithm was selected to carry out the word recognition task. The algorithm was to be tailored to the specific problems of the acoustic processor.

Scope

The target vocabulary was to be small, less than 100 words. The vocabulary was to support a minimal cockpit environment, consisting of the digits and a few command words. Table II is a listing of the selected vocabulary.

The phoneme strings to be parsed to words were to be constructed from a set of 39 universal phoneme prototypes. These phoneme prototypes were to be under revision by another researcher. Table III is a list of the initial phonemes used.

The speakers used for testing the system were to be principally this investigator (Baker) and other American males deemed useful for inclusion in the testing. Speech to be used was to be confined to digit strings such as ZERO ONE

	Table II abulary Se	t
ZERO	8.	SEVEN
ONE	9.	EIGHT
TWO	10.	NINE
THREE	11.	CCIP

5.	FOUR	12.	SET
6.	FIVE	13.	ENTER

1.

2.

3.

4.

7. SIX 14. FREQUENCY 15. THREAT

THREE FIVE NINE with a maximum time constraint of three seconds of speech. The time constraint was imposed by an inability of the available hardware to sample and store more than three seconds of speech.

Recognition was to be achieved by a relatively simple pattern matching process since it was believed that little linguistic/syntactic information would be available in the limited vocabulary.

Table III
Phoneme set

ł					
Compt Symbo		Key Word	Comp Symbo	uter ol	Key word
1.	PX	рау	21.	JX	уou
2.	вх	<u>b</u> e	22.	LX	let
3.	TX	<u>t</u> o	23.	RX	<u>r</u> ead
4.	DX	<u>d</u> ay	24.	EE	<u>e</u> ve
5.	KX	<u>k</u> ey	25.	IX	it
6.	GX	go	26.	EI	h <u>a</u> te
7.	нх	he	27.	ЕН	m <u>e</u> t
8.	H@	a <u>h</u> ead	28.	AE	<u>a</u> t
9.	FX	for	29.	AX	<u>a</u> sk
10.	VX	<u>v</u> ote	30.	AA	f <u>a</u> ther
11.	TH	<u>th</u> in	31.	AH	n <u>o</u> t
12.	DH	<u>th</u> en	32.	AW	<u>a</u> ll
13.	SX	<u>s</u> ee	33.	OU	<u>o</u> bey
14.	ZX	z 00	34.	UX	f <u>oo</u> t
15.	SH	<u>sh</u> e	35.	บบ	b <u>oo</u> t
16.	ZH	a <u>z</u> ure	36.	UH	<u>п</u> р
17.	MX	m e	37.	UH	<u>a</u> bout
18.	NX	no	38.	ER	w <u>o</u> rd
19.	NG	si <u>ng</u>	39.	XX	NA
20.	WX	<u>ж</u> е			
ľ					

Approach

The research divides naturally into the categories of: software development, system training, and system testing. The software to be developed included a system to execute the pattern matching algorithm and an editor to alter the word templates.

Ultimately, two separate systems were constructed with one being discarded. The final system was a state machine capable of parsing arbitrary patterns within the memory limits of the host processor. The actual software for executing a state machine is quite simple, however, the primary work of implementation lies in determining the state The main result of the system training change rules. becomes embodied in the memory contents representing the system states. The minimal software necessary to execute the state machine resulted in a program which combined the execution algorithm into the same program module as the This resulted in an interactive system which was editor. easy to debug and train. The combined editor and execution module permitted observation and editing of the executing state machine.

II. The Acoustic Processor

<u>Hardware</u>

The acoustic processor, which is the major component of this speech recognition system, is a powerful, generalized assembly of hardware and software. Figure 1 outlines in block diagram form the major components of the acoustic processor.

The computer which performs the signal acquisition and processing is relatively straight forward in its construction. Its design is a product of cooperative work between the AFIT faculty, principally Captain Larry Kizer, and various AFIT students, most notably Captains George Beasley and Dan Fredahl. Beasley and Fredahl (Ref 6) are responsible for the I/O processor embodied in the Cromemco computer.

The signal processor is composed of three interconnected computers with limited communications between the processors. They do not constitute a network in the sense that all the processors communicate directly with one another. The Nova 2 is the central processor in that it communicates with the Eclipse and the Cromemco, but the Eclipse and Cromemco do not communicate with each other.

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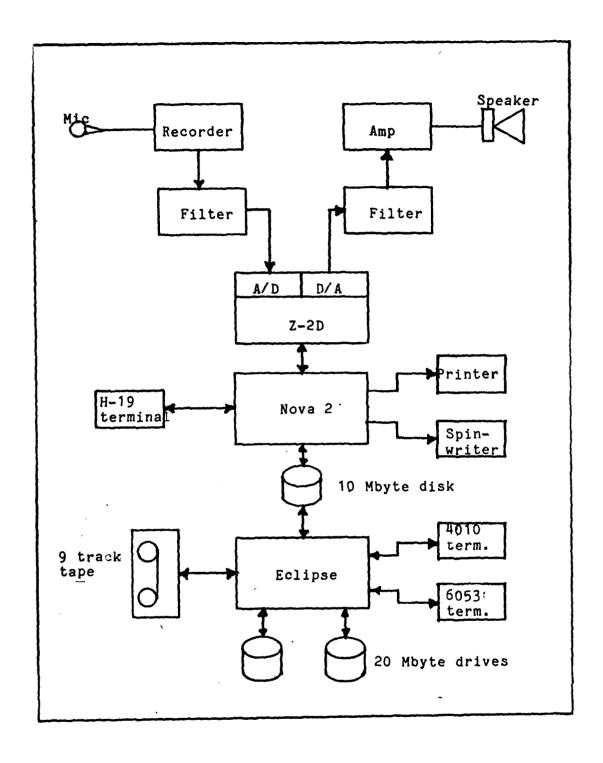


Figure 1. Acoustic processor block diagram

The signal processor can process both audio and video Cromemco Z-2D Z - 80information. The is based microcomputer containing 12 bit digital converters, both A-D (analog to digital) and D-A (digital to analog). It also contains 4 bit video A-D and D-A converters for video input The video converters contain their own high and output. speed memory but the memory for the audio digital converters is the 64K byte memory of the Cromemco computer. information is moved in and out of the Cromemco memory through a DMA (direct memory addressing) process. byte Cromemco memory is the constraint which limits the processing of speech to utterances less than three seconds We customarily process speech at a digitization rate of 8KHz. Since 2 bytes are required for each 12 bit word passed to or from the audio digital converters, 16 kilobytes of memory are used each second. Leaving room in memory for the operating system results in a practical time limit of approximately three seconds of speech at 8KHz. Cromemco is interfaced to the Nova 2 through a high data rate buss for speech storage. Speech segments may be digitized, stored, and played back through the digital to analog converters very conveniently.

The Data General Eclipse computer is a relatively powerful general purpose mini-computer. It is interfaced to the Data General Nova 2 and three hard disk drives. drive shared with the Nova 2 is of 10 megabyte capacity and the other drives are 20 megabyte capacity. The 10 megabyte disk drive is of the common figuration which provides a 5 Mbyte 'fixed' disk and a 5 Mbyte removeable disk. The fixed disk is fixed only in the sense that it is not to be removed from the drive by the computer user. The other disk may be removed to permit the transport or protection of data on a disk. The Eclipse also has a 9 track tape drive used primarily for archival storage of data and programs. two Data General computers, sharing a common disk drive, communicate with each other through an inter-processor communication buss. The inter-processor buss functions primarily to prevent the two computers from accessing the common disk drive simultaneously. The normal mode of passing data between the two computers is through files placed on the common disk drive. One computer writes a file which may then be accessed and processed by the other computer.

The Eclipse operates two CRT (cathode ray tube) terminals. One of them is a standard text display terminal but the other is a Tektronix 4010 vector graphics display terminal. The graphics terminal serves an important

function in speech work in that a program has been developed which permits it to display speech spectrograms. The current function of the system permits an operator to designate a section of a speech spectrogram on the graphics terminal and then hear the exact segment of speech responsible for producing the spectrogram. The segments of speech played back may have a duration ranging from a few milliseconds up to approximately three seconds.

The Nova 2 provides system I/O for both the Data General computers. It controls both printers currently connected to the system. The primary printer is a Printronix high speed dot matrix printer. This allows the of relatively high quality graphics for scene output recognition work, as well as the printing of sprectrograms. The Printronix also performs the mundane task of listing data and programs. The second printer connected to the Nova 2 is a high quality daisy wheel printer.

Signal Handling

The speech signals may be acquired from a microphone and digitized immediately, or recorded on tape and then digitized. In the direct acquistion mode, the tape recorder electronics function as a preamplifier for the microphone. The amplified signal is passed through a Rockland active

filter. This filter is capable of 8 pole high pass and low pass function over a wide frequency range but for speech processing it is set as an 8 pole low pass Butterworth filter with a 3db cutoff at 4KHz. An 8 pole filter results in an attenuation slope of 48 db per octave. Following the filter, the signal is immediately digitized and stored. For speech recognition, the signals receive a 6db per octave logarithmic pre-emphasis beginning at 500 Hz. This pre-emphasis is not a hardware function; it is accomplished by software.

The signal processing system is configured to allow very convenient acquisition of speech signals. Acquiring a speech sample requires that the operator execute a program which supervises the process. The supervisory program currently used is AUDIOHIST, written by Captain Paul Finkes. When executed, this program requests a file name for the speech sample and allows the operator to choose between input and output. Upon selecting input, AUDIOHIST allocates file space and sets up the Cromemco I/O processor to accept a speech input. When all is ready, a lighted button, convenient to the control console and microphone, begins flashing. Depressing the button immediately starts the analog to digital converter. The normal action is to press the button and begin speaking. Upon reaching the time limit of three seconds, the lighted button signals the end of input. AUDIOHIST is available to allow an immediate scan of the acquired samples. It provides a condensed listing of the maximum input voltages attained, with an approximate indication of when they occurred. This permits a convenient check to ascertain that the audio gain was within reasonable limits. The general goal in checking speech input voltages is to attain as high a level as possible without saturating the A-D (analog to digital) converters. AUDIOHIST also permits immediate playback of the acquired speech sample. This is done routinely to insure that parts of words are not missed.

Software

The software and associated algorithms naturally constitute the important functional parts of the speech recognition system. The current software design is largely the work of Lt. Mark Felkey (Ref 2) but has recently been modified extensively by Lt. Karl Seelandt.

Felkey's algorithm in a strong sense follows from the work of Potter, Kopp, and Green (Ref 3) in their work on speech spectrograms, published in 1947. They made the observation that persons familiar with speech spectrograms were able to read the spectrograms in much the same sense that we read text. They also noted that spectrograms from different speakers were similar for the same utterance.

This similarity between different speakers provides the justification for Felkey's approach of accomplishing phoneme recognition based on spectrographic data (Ref 2:12).

In processing an analog speech file for the first step is the production of a spectrogram of the entire utterance. Each vertical column of the spectrogram, Figure 2, represents 64 digitized samples or 8 milliseconds of analog speech. Performing a DFT (discrete Fourier transform) on the data results in a 32 point DFT. Only the magnitude of the spectral components is retained for further processing. The entire file is processed 64 samples at a time, using rectangular windowing. The resulting spectrogram contains frequency magnitude information from DC to 4Khz, with frequency samples at 125 Hz intervals. The choice of 64 samples for each DFT is very much empirical.

After the spectrogram is available, a correlation program determines which phoneme is the best possible choice for each column of the spectrogram. The phoneme templates used in the correlation process are prepared by listening to short segments of a speech file. A program developed by Seelandt permits veiwing a spectrogram on the 4010 graphics terminal and then designating a short sequence to be played back over the system speakers. The spectrogram is not



Figure 2. Spectrogram of the word "one" uttered in (a) connected speech and (b) from isolated speech.

played back but rather the segment of digitized speech which produced the spectrogram is played back. This permits the isolation of potentially very short, well defined phoneme segments to use as templates. The templates are placed in a file which the correlation routine compares against the spectrogram, one column at a time. Figure 3 is a spectrogram of several of the phoneme templates. phoneme template is 5 columns wide or 40 milliseconds time duration. The correlation process begins at the fifth column of the speech spectrogram. Each phoneme template, which is five columns wide, is compared to columns 1 through 5 of the speech spectrogram. The results are tabulated and ranked then the process is repeated by comparing each phoneme template to columns 2 thru 6 of the spectrogram. A tabulation is produced for each 8 milliseconds of speech until the entire speech file has been been through the correlation process.

Felkey produced spectrograms based upon several different sample sizes. He judged the 64 sample spectrogram to be the one which produced maximum intelligibility for a human, yet which reached a 'good' compromise between time and frequency resolution (Ref 2:15). (An unfortunate feature of the DFT is that as frequency resolution increases, time resolution decreases and vice versa.) The

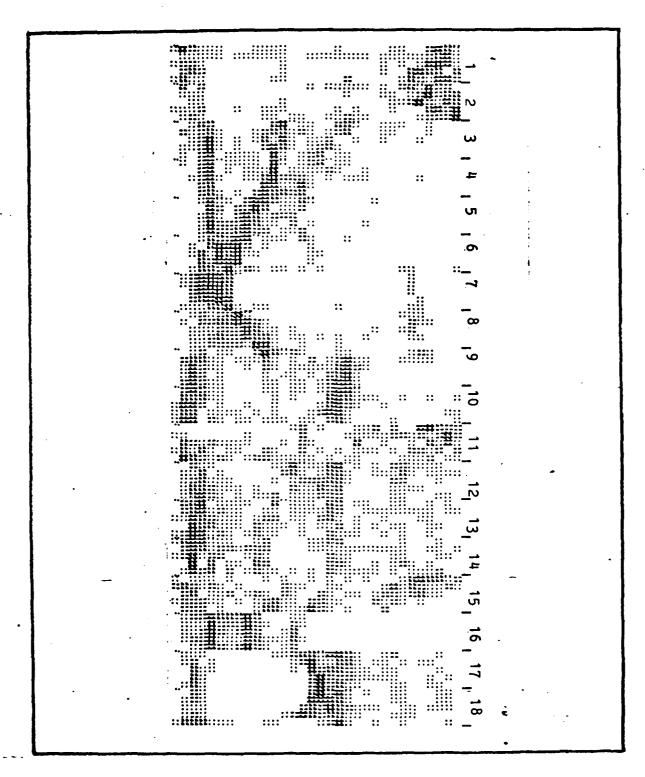


Figure 3. Phoneme Templates

'good' compromise is one which allows resolution of the shortest phonemes which Felkey observed to occur in the speech samples he examined. In current work with the system, we have seen phonemes which extended over only two columns of the spectrogram. This represents a time period of 16 milliseconds since each column of the spectrogram represents 8 milliseconds of speech. The short phonemes were observed on the very beginning of the 't' in the word 'two'.

Using short phoneme templates as this algorithm does appears to obviate the need to account for variations in speech rate. Since a phoneme selection is made every 8 milliseconds, the normal output of this program produces multiple matches over time. For instance, a phoneme lasting 100 milliseconds in the speech file will result in the phoneme recogniton algorithm producing 12 or 13 output matches. If someone speaks slowly and stretches phonemes, the Felkey algorithm simply indicates that multiple occurences of the phonemes are joined to produce the longer utterance.

III. The State Machine

Introduction

The recognition algorithm studied in this thesis is implemented as a finite state machine. The primary motivation for choosing such an implementation was a recognition that the more successful of the ARPA SUR projects were based upon state machines. The HARPY machine, constructed as part of the ARPA SUR project, was probably the most successful speech recognition machine yet built; it was a finite state machine (Ref. 1). HARPY provided the initial germ of idea for this algorithm. A review of the properties of a finite state machine resulted in deciding that a finite state machine would serve my purposes well.

Reasons for Choice

First, a state machine is, or may be, a very powerful algorIthmic device. Every digital computer is conceptually a finite state machine. Though even a modest digital computer is capable of a very large number of states, it is nevertheless, a finite state machine (Ref 5:63).

Second, the operation of a state machine is generally quite easy to visualize. This may be a personal bias, but I believed that the ability to visualize the operation of the

recognition algorithm was quite important. I am much more comfortable with state diagrams than computer program flowcharts and was thus strongly biased to choose a state machine.

Third, a state machine is intrinsically capable of determining order. The first recognition algorithm attempted under this thesis was not designed to consider the order in which phonemes occurred. Work performed with that algorithm, however, showed that recognizing the order of phonemes was necessary even in the small vocabulary which we are using. This was an unanticipated result and stemmed from the nature of the errors made by the specific acoutic analyzer being developed in the AFIT signal processing laboratory.

Fourth, a state machine is directly capable of timing events. In the algorithm under discussion, a state machine would accomplish timing by counting events. The actual time duration of each event is measured by the analog to digital convertor in the acoustic processor. Phoneme duration was not considered important for this application, but the inherent timing capability which the algorithm proffered gave some hope of simple correction should the assumption prove wrong.

Fifth, a state machine simulation appeared to be one of the few software implementations capable of being "rolled into and out of" a computer memory and modified without recompiling. No research was performed to find other techniques but the capability is unique within my experience. What this really means is that a state machine simulation was constructed in which the state transition information was held in a data file. The data file was separate from the program which exercised the state machine. We were thus able to modify the state transition information quickly, easily and without recompilation of the program.

What is a State Machine?

Let's begin by considering a state machine as a concept rather than a physical device. Viewing it as a concept is particularly appropriate here, since the state machine implementation is in software.

A digital system may be modeled as a state machine. In the model, a digital system is regarded as a system which moves from one state to another in discrete steps. Each transition or step is determined by the inputs to the system as well as the current state of the system. At each transition, the system may output information. In a software implementation of a state machine, each state is

represented by a series of ones and zeroes stored in memory (Ref 5:63).

To accomplish useful tasks, a state machine must have a defined start state. In terms of a computer program, the start state corresponds to the initialization process which any program must accomplish. Certain memory locations or registers must be set to zero, other storage locations must be set to other predefined values. If this initialization is not accomplished, the program will produce essentially random results based upon the content of the storage locations.

Implementation

The word recognition algorithms discussed were implemented on an Apple II computer in UCSD Pascal. The Apple had 64 kilobytes of random access memory, two disk drives allowing online storage of 286 kilobytes of programs and data, a printer and serial interface devices which permitted direct communication with the Nova computer in the acoustic processor.

The Apple, using Pascal, is a powerful, flexible machine which permits simple access to the inner workings of various programs and processes. Pascal implemented on the Apple cannot be relegated to the status of a mere subset of

the standard Pascal language. It is missing only significant capability and that is the ability to pass a procedure name as a formal parameter to another procedure or function. When compared to the Pascal implementation on the Cyber 6600, used by AFIT, the UCSD Pascal is actually more powerful in certain respects. The UCSD Pascal will accomdate set definitions with 512 members. The Cyber is limited to approximately 54 set members. These facts are normally of little significance but one algorithm considered and tried briefly required large sets. The language on the Apple was actually more suitable and attractive than the Cyber. Certainly the Cyber executes code much faster has greater arithmetic precision, but neither of these capabilities was of any value to this project. The convenience and accessability of the Apple overshadowed the power of the Cyber or other computers available.

The state machine, as implemented, is little more than an array of data with a mechanism to accept inputs and determine the next position in the array. A position in the array corresponds to the current state of the machine. The state machine constructed was capable of 256 states, and a single input. The machine was implemented with 256 states because this number of states may be addressed with a single byte, or 8 bits, of information. Exceeding 256 states would immediately double memory storage requirements. The input

accepts 40 different values, corresponding to 40 different phonemes. Any state is capable of being an output state. An output state is one in which the algorithm determines that a word has been found. The normal output condition is to indicate "no output" at each state transition. However, when the machine has stepped through a sequence of states, indicating detection of a word, that word is typed out.

Data Structure

Since the state machine implementation and algorithm constitute a major goal of this thesis, it is appropriate to discuss the data structure and and recognition strategy.

Figure 4 is a diagramatical representation of the state machine data structure. In the terminology of Pascal, this is a record of arrays. Records are an important structure within Pascal, they are responsible for much of the power of the language. A record is a high level data structure which the programmer defines. The structure represented in Figure 4 is composed of three different arrays. The power of a record resides in the ability of the system to treat the record as a single entity. The arrays which compose the record are:

A two dimensional array of integers with values
 from 0 to 255 with subscript ranges from 0 to 255

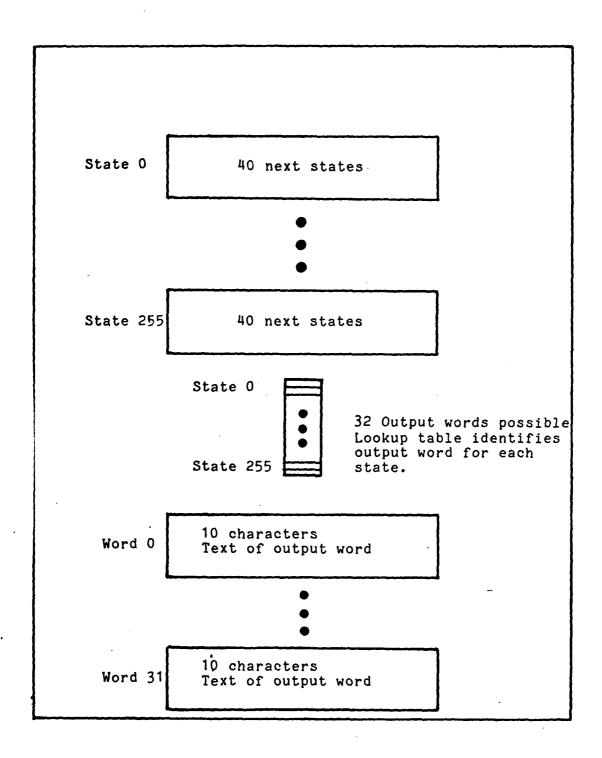


Figure 4. State machine data structure

and 0 to 39.

- 2. An array subscripted from 0 to 255 taking integer values from 0 to 31.
- 3. A two dimensional array with subscripts of 0 to 40 and 1 to 10 with each cell of the array being a character variable.

The actual arrangement of the arrays is insignificant. What is important is the ability to define such a structure, place data within the structure and then write the data into permanent storage as binary image of the structure as it existed in memory. The system treats the record as a entity and reads the record from storage or writes it to storage as a single entity. The record as described in Figure 4 (and as implemented) occupies approximately 10,000 bytes of storage. The programmer is naturally allowed access to the individual items of the record to read or modify them.

Recognition Strategy

Figure 5 is a state transition diagram to recognize the word zero. Let's discuss it in some detail. The discussion will illuminate the inner workings of the state machine and reveal the recognition strategy.

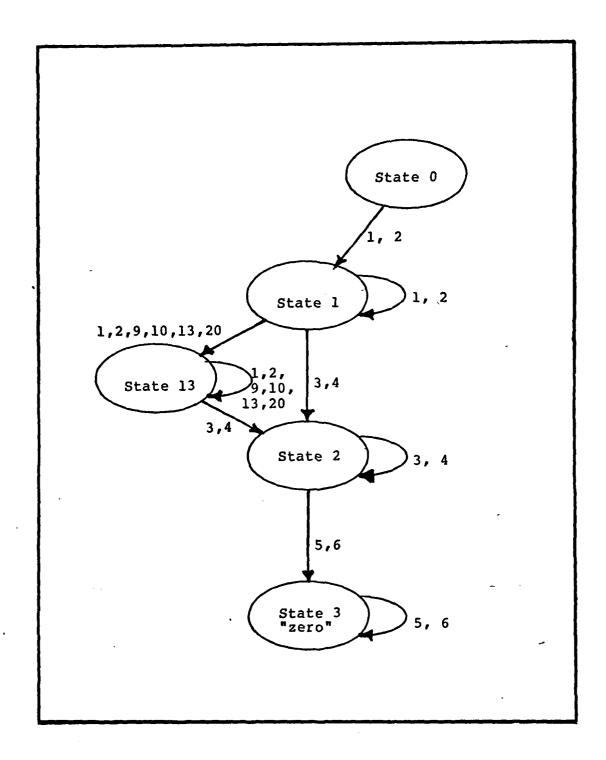


Figure 5. State transition diagram of word "zero"

Each circle represents a state and the number within the circle is the state number. As previously mentioned, state 0 is the starting point for all recognition attempts. There are many paths leading out of state 0 but we will only discuss two of them for the moment. The phonemes which will produce a transition from one state to another are listed between states. The phonemes are the numbers beside each State 1 has an arrow which loops back to state 1. arrow. Listed beside that arrow are phonemes 1 and 2. The meaning is that so long as the input phoneme is 1 or 2, the machine will remain in state 1. Once the machine has reached state 1, an occurrance of phonemes 9, 10, 13, or 20 will change the state to state 13. The phonemes listed are explicit state changes. Another state change which is implied, but listed nowhere, is that upon reaching state 1, if any phoneme occurs other than 1, 2, 3, 4, 9, 10, 13, or 20, the the machine transitions to state 0. This is both a strong point and a weakness of the chosen recognition strategy. It is a strength of the algorithm in the sense that it appears highly unlikely that this algorithm will confuse one word for another. If one single phoneme occurs which is not within the experience of the system, the decision is that the most recent phoneme string is unrecognizable. weakness, of course, lies in the intolerance of the system to deviations from patterns on which it has been trained.

PM 1

The term "within the experience of the system" requires explanation. It means that the system must have been trained to recognize the string under consideration. possible deviation from the expected path to a word must have been defined. In practical terms, this requires that effort be devoted to training the recognition great algorithm. All tolerated deviations from the expected state path must be explicitly determined and entered into the state transition matrix. The training strategy adopted requires that the trainer examine the state transition diagram for a word an integrate new word templates into the The expectation is that as existing structure. patterns are processed and entered into the state transition table, few changes will be made as each new word is processed. I'm not speaking of "new" in the sense of words not previously in the vocabularly. It means new utterances of words already in the vocabularly.

Returning to the explanation of Figure 5, phonemes 1 and 2 are the "z" sounds of zero. Phonemes 3 and 4 are the "er" and phonemes 5 and 6 the "oh" sound to terminate the word zero. All other phonemes listed are observed errors and deviations from the expected phoneme string.

Obviously, this technique is highly dependent upon the function of the acoustic processor and the speaker. It does

appear possible, though, to allow for speaker variations by simply allowing new paths to a word. Certain types of deviations are allowed but if the deviations are not within the experience of the system, the state machine returns to state 0 and begins attempting to parse a new word.

IV. Training the Recognizer

Training the word recognizer is at best tedious and time consuming. Furthermore, it does not appear that the process is amenable to automation. The reason for this is simple. Automating the training process requires a more capable recognizer to accomplish the training than is required to carry out the word recognition process.

The training process which I have followed is very time comsumptive but minimizes memory use in a general sense. Most important, the training process appears capable of producing generalized word recognition templates. An automatic training algorithm is a desireable goal but it is not clear that one can be constructed. It appears that an automatic trainer capable of generalizing the word templates would be more complex than the word recognition algorithms.

The training process begins with the state machine initialized so that all possible state transitions result in a jump to state 0. A jump to state 0 is the required action when the input phoneme results in a string which the state machine has not been trained to recognize. We shall begin with a description of training the machine to recognize the word "zero". However, before we begin the description, several details of the acoustic analyzer must be

illuminated.

The phoneme set used for this work consists of phonemes taken from the words "zero", "one", "two", "three", "four", "five", and "six". There are 28 phonemes in this set. They are listed in Table IV. The keyword listed for each sound is not appropriately marked in a phonetic spelling sense. The underlined portion of each word is an approximation of where the phoneme was obtained. Note that the phonemes are duplicated in several instances. The phoneme "ou" appears in three different places as phoneme numbers 5, 6, and 20, for instance. In general, when a phoneme appears with a number as part of the name, as in UU1, the lower numbered phoneme represents a beginning segment of the phoneme and the higher numbered phoneme a later segment of the same sound.

Very briefly, the steps required to train the state machine to recognize a word are:

- 1. Obtain a digitized speech file.
- 2. Process the speech file to obtain a spectrogram.
- 3. Process the speech file to obtain phoneme recognition results from the acoustic analyzer.

Table IV												
Computer Symbol		Phoneme Key Word		uter ol	Key Word							
1.	Z1	zero	15.	TH	<u>th</u> ree							
2.	Z2	<u>z</u> ero	16.	R	th <u>r</u> ee							
3.	ER1	z <u>er</u> o	17.	EE1	thr <u>ee</u>							
4.	ER2	z <u>er</u> o	18.	EE2	thr <u>ee</u>							
5.	001	zer <u>o</u>	19.	F	<u>f</u> our							
6.	005	zer <u>o</u>	20.	on	f <u>ou</u> r							
7.	WO	<u>o</u> ne	21.	UH	five							
8.	AH	<u>o</u> ne	22.	IE	f <u>i</u> ve							
9.	N1	o <u>n</u> e	23.	V	fi <u>v</u> e							
10.	N2	o <u>n</u> e	24.	S1	<u>s</u> ix							
11.	T	<u>t</u> wo	25.	S2	<u>s</u> ix							
12.	UU1	t <u>wo</u>	26.	IX1	s <u>ix</u>							
13.	บบ2	t <u>wo</u>	27.	IX2	s <u>ix</u>							
14.	UU3	t <u>wo</u>	28.	к -	si <u>x</u>							

4. Examine the spectrogram to determine the time at which each word in the speech file begins and ends.

- 5. Mark the phoneme recognition results with the data obtained in step 4.
- 6. Determine which phonemes in the phoneme recognition results represent "correct" phonemes and which are errors from the acoustic analyzer. This step is simply a best guess and may later be revised.
- 7. Form a state diagram based upon the results of step 6.
- 8. Enter the state diagram information into the state machine.

"Zero" is ideally composed of the phonemes "z",
"er", and "ou". Table V is a list of phonemes, for the word
"zero", which have been extracted from a larger list
produced by the acoustic analyzer in processing a speech
file containing the words "zero", "one", "two", "three", and
"four". The phonemes listed are those which the acoustic
analyzer indicated as having the highest correlation for
each time segment of the speech file. The numbers 48
through 78 indicate the time slice number since the
beginning of the speech file. Recall that each time slice
represents 8 milliseconds of speech.

Table V Phonemes of word "zero" Phoneme Segment Phoneme Phoneme Segment Phoneme Number Number Number Number Name Name 48 2 **Z2** 64 3 ER1 2 **Z2** ER2 49 65 2 **Z2** 66 ER2 50 67 ER2 51 Z 1 52 Z 1 68 ER2 **Z** 1 69 ER2 53 54 **Z1** 70 ER2 1 55 2 **Z.2** 71 ER2 56 1 21 72 13 บบ2 2 บบ2 57 **Z2** 73 13 58 2 **Z2** 74 16 R **OU1** 59 1 **Z1** 75 5 **UU2** 60 13 76 5 0U1 61 **UU2** 16 13 77 R 62 13 UU2 78 16 R 63 3 ER1

Based upon the information in Table V, we can form the state diagram of Figure 6. Note that the first phoneme of Table V is "Z2" or phoneme 2. Figure 6, the state diagram,

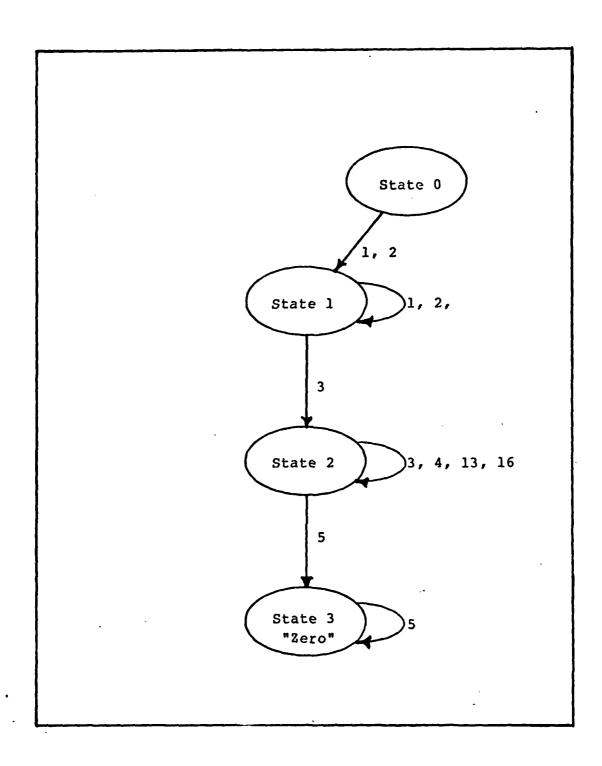


Figure 6. State transitions for "zero"

has been chosen so that Z1 or Z2 will produce a transition to state 1, the start of the "zero" string. Upon reaching state 1, input phonemes 1, 2, and 13 will yield a next state of state 1. Phoneme 13, occurring at time segments 60, 61, and 62 of Table V, is an error from the acoustic processor. However, because we have now seen this phoneme occurring in a "zero" string, we shall include it in the "experience" of the state machine. This is accomplished simply programming the state machine so that while in state 1, input phoneme 13 produces a next state of state 1. state 1, an input of "ER", which is phoneme 3 or 4, will cause a transition to state 2. While in state 2, the machine waits for the occurrence of the final phoneme in the "zero" string, which is an "OU", phoneme 5. At time segments 72, 73, and 74 observe the errors. Before completing the "zero" string, phonemes 13 and 16 appear. We choose to incorporate these by remaining in state 2 and waiting for phoneme 5. When phoneme 5 occurs, the machine transitions to state 3 and produces an output to indicate that the word "zero" has been found.

Several assumptions are implicit in this procedure used to train the machine. The first assumption is that when the machine reaches state 3, the word "zero" has occurred.

Given the nature of the machine this is not necessarily a true assumption. It is possible to step from state 0 to state 3 with only three inputs. Three inputs would only represent 24 milliseconds of speech. Obviously, no one speaks the word "zero" in that length of time. However, we have not observed the sequence to occur other than in the word "zero". The best we can say is that it appears to be a reasonable assumption to believe that upon reaching state 3, the word "zero" has been processed.

Another potentially nettlesome assumption is the error processing. Is it reasonable to simply wait for the correct phonemes to occur when we are observing "errors"? This problem is eliminated by redefining the "errors". The "error" phonemes are not really errors within the context of recognizing the phoneme string as a word. They are simply phonemes which have been observed to occur while processing the word. From the viewpoint of a human observer, the phonemes have no place in the word. However, since we have observed that they occur and do lead to a correct parsing of the word, they will be incorporated in the knowledge of the state machine.

The final step in the training process is to incorporate the state diagram of Figure 6 into the state transition table of the state machine.

Incorporating the second, third, and successive occurences of "zero" into the state table will follow much the same process. Where possible, new error phonemes should be handled by self looping, as was done for phonemes 13 and 16 on states 1 and 2. It is our belief that this procedure generalize the recognition process so that new occurrences of the word "zero" will eventually produce no changes in the state transition table. If a new path to state 3 were created for each occurrence of "zero" which differed from previous "zeros", we would quickly consume all available memory and simply have several paths which lead to By attempting to incorporate all possible state 3. variations into one path, the path becomes very generalized and permits variations within the experience of the system, yet forces certain criteria to be met.

V. Results and Conclusion

The primary result of this thesis is an algorithm described in previous chapters which may be a useful continuous speech word recognizer. The technique described is well adapted to the data produced by the AFIT acoustic analyzer and appears to have a high potential of functioning to recognize a small vocabulary. For several reasons the technique does not appear capable of handling a large vocabulary. The foremost reason is simply the problem of finding sufficient differences between words. For the small vocabulary considered for this project, the differences between the words are great enough as to present no problem. As the vocabulary increases, it is safe to say that the distance between various words will decrease and the algorithm will encounter difficulty parsing the words. major difficulty in a larger vocabulary will be the method selected to handle error phonemes.

What can be said of this algorithm is that it successfully parses each utterance which has been incorporated into the state transition table. A total of fifteen utterances have been incorporated for the words "zero" through "six". No utterance has been recognized before the state machine was trained to recognize that

particular utterance. It is understood that a machine which must be trained for each individual pattern it is to recognize is useless. However, it appears that the changes necessary to incorporate each utterance are decreasing as the incorporated experience grows.

VI. Recommendations

The most important recommendation which may be made to anyone contemplating continuing this project or another of similar nature is that the project should not be considered unless the acoustic analyzer is under the control of the individual attempting word recognition. The recognition work attempted under this thesis has done little more than verify that the software functions as intended. It does not suffice as as a test of the recognition algorithm.

The reasons for the recommendation are relatively simple. If another individual is extending the acoustic analyzer, he or she must be free to make changes to suit their goals of phoneme recognition. In general, the goals of phoneme recognition and word recognition are compatible. Unfortunately, the word recognition goal requires a stable, copious output from the acoustic analyzer to build and verify the word recognition process. Neither goal of stable or copious output will be attained while someone else is modifying the acoustic analyzer.

The next recommendation is incorporate more utterances into the state transition table described and verify or disprove the algorithm described. I estimate that the word recognizer is likely to begin functioning before 10

utterances of each word have been incorporated in the state table. That is, begin functioning in the sense that words will be recognized based on the templates already incorporated in the state table.

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APPENDIX A

COMPUTER PROGRAM NOTES

The program described is a single input state machine of 256 states. Each state has 40 possible next states, based upon 40 different input phonemes.

Combined in one program are all the tools needed to execute the state machine, to edit and change any information contained in the state table, to alter the output information from the machine and to transfer the next state tables to the printer in a readable form.

Upon executing the program, there is a prompt line which is "Which state machine?" The proper response is a single character such as "1" or "A", etc. The character will be incorporated into the file name, STMACHx.DATA, where x is the input character. The program will attempt to open If the file is opened successfully, the program that file. continues, otherwise the prompt is repated. Entering a period will cause the program to ignore any data files and continue but the memory locations reserved for the state machine will all be set to zero. This is the proper condition for defining a new state machine, you might say a "clean slate". The previously described action is accomplished by the init procedure.

When the state machine has been loaded to memory, a prompt line appears with the options, "E, P, Q, X, V, W".

Typing "E", calls the editor. The editor allows the user to freely examine and modify the contents of the state machine. The screen display is similar to Figure 9. The numbers 1 through 40 are the possible input phonemes. The other information, mostly zeroes, are the next states. While in this state, input phoneme 2 will cause a jump to state 1, input phoneme 4 will cause a jump to state 25. The cursor may be moved from one phoneme to another with the Apple Pascal editor cursor movement commands. The move keys are cntrl-O, cntrl-L, right arrow, left arrow, and return. Cntrl-O moves the cursor up and cntrl-L moves the cursor down. The phoneme to the immediate right of the cursor may be modified by by entering digits. You may terminate the entry with a return or a cursor move. The editor may also be entered from either execution mode.

Typing "P", for print, will result in the program asking which states are to be printed. If it were told to print states 1 through 25, the result would be an output similar to Figure 9, repeated 25 times. Three next state tables are printed on each page.

Typing Q will start the process of exiting the program. You will be asked whether the state machine in memory should be saved. If no data is to be saved on disk, the program may be exited by typing a series of "Q's".

1 0	2	3	4 25	5 3	6 0	7 0	8 0	9	10 0
11	12	13	14	15	16	17	18	19	20
0	0		0	0	0	0	0	0	0
21	22	23	24	25	26	27	28	29	30
0	0	0	0	0	0	0	0	0	0
31	32	33	34	35	36	37	38	39	40
0	0	0	0	0	0	0	0	0	0
State: Output	3 word	is	[no ou	tput]				

Figure 7. Next state screen display

Typing "X" will put the machine into an execution mode in which the only display is the output words and input phonemes. No next state table is displayed.

Typing "V" places the program in an execution mode in which the next state table is constantly displayed and updated for each state transition. For either execution mode the phoneme inputs may come from the keyboard or a file.

Typing "W" allows entry of the text of the output words. A word is displayed for each state while editing or

executing. The normal output is "no output".

The user may exit the program by typing a aeries of "q's". The number varies depending upon the exact status of the program. If you are using the editor from inside the state machine execution module, four "q's" may be required to get out of the program.

APPENDIX B
COMPUTER PROGRAM

```
{ $1 printer:}
program state;
uses io. applestuff;
   const
      starray = true;
      outarray= false;
      maxphons=39;
      stline=16;
      wordline=17;
      esc=1;
   type
      stmach = record
                  nutst:packed array(0...255,0..mamphons) of 0...255;
                  word :packed array[0..255] of 0..31;
                  wtext:packed array(0..40,1..10) of char;
               end;
      numtype = '0'..'9';
      numsetype = set of numtype;
   Val
      sm:stmach;
      lines:integer;
      zmode:boolean;
      numset:numsetype;
      name:string;
      ch:char;
      visible quit: boolean;
      eoline:char;
      phonum, edstate: integer;
      mpos:packed array[0..mamphons] of 0..mamphons;
      ypos:packed array[0..maxphons] of 0..maxphons;
      inphons:text;
```

```
Init performs the task of initializing
                                                      global
         variables and and setting up the state machine.
procedure init;
VAI
  ch:char;
  result: integer,
   smfile: file of stmach;
  i,j,pnum:integer;
begin
  visible:=false;
   quit:=false;
   edstate:=0;eoline:=chr(13);
  numset:=['0'..'9'];
         Go to the console and ask which state machine file
         is to be used. The state machine files are all
         identified as STMACHx.DATA. The "x" is any single
         character other than a "." which the Pascal system
         will accept. Typing a period in response to the
         prompt "Which state machine" will result in
         initializing the state machine so that all next
         states are State 0.
   repeat
      writeIn('which state machine? ');
      sead(keyboard,ch);
      if eoln(keyboard) then ch:='0';
      name:='stmach .data';
      name[7]:=ch;
      (#$i=#)
      reset(smfile,name);
      (*$i+*)
      result: =ioresult;
      if (result(>0) and (ch(>'.') then
         writeln(name,' not on prefixed volume');
   until (result=0) or (ch='.');
```

A LON

```
Now read in the state machine or set every thing to
3
  if resuit=0 then
    begin
      get(smfile);
      sm:=smfileA;
  else fillchar(sm, siteof(sm), chr(0));
  pnum: =-1;
(
Set up the arrays, xpos and ypos, with the x y
       co-ordinates where the next state will be displayed
       for each input phoneme.
......
3
  for i:=0 to 3 do
    begin
      for j:=1 to 10 do
         begin
           pnum:=pnum+1;
           xposipnum := (j-1)*4;
           ypos(pnuml:=i*3 + 4;
         end;
    end;
end:
```

```
Procedure savesm saves the state machine by writing
         it to a data file. It defaults to using the name of
         the state machine which was read from disk at
         initialization. If it is desired to save the state
         machine under a new name, simply type a character.
         The character will be incorporated into the data
         file name as STMACHx. DATA where "x" is the character
         entered.
procedure savesm;
  smfile: file of stmach;
  result: integer;
  quit:boolean;
  ch, chi: char;
begin
  quit:=false;
  repeat
     repeat
        clear;
        writeln('save the file as ',name,'?');
        writeIn('(return) to use this name or enter char');
        read(keyboard, ch);
        if not coln(keyboard) then name[7]:=ch;
        writeln('saving as ',name);
        writeIn('ok? y/n/q');
        ch1:=getchar(['q','Q','y','Y','n','N']);
        case chi of
           'q','Q': begin
                      quit:=true;
                   end;
        end;
               (*case*)
     until (chi='Y') or (chi='y') or quit;
     if not quit then
        begin
           (*$1-*)
           rewrite(smfile, name);
           smfileA:=sm;
           put(smfile);
           result:=ioresult;
           close(smfile,lock);
           (*$i+*)
```

end;

until quit or (result=0); end:

end;

```
Procedure putsorn paints the screen. This is the
         primary display procedure while operating in the
         visible mode.
                        In the visible mode, the screen is
         updated for each state change. The screen display
         presents the next state for each possible input
         phoneme. In addition, the screen displays the
         output for the current state.
procedure pntscrn;
  j,phonumb:integer;
begin
  clear;
  topline;
  gotoxy(0,3);
         The first for loop involving phonumb writes a line
         which labels the next states. On the first pass
         through the "for j:=1" loop the labels are 1 through
         10. On the next pass, 11 through 20, etc. Fourty
         next states will be displayed and labeled.
         second for loop involving phonumb actually writes
         the next states.
   for j:=1 to 4 do
     begin
        for phonumb:=1+(j-1)*10 to j*10 do
           write(phonumb:4);
           writeln();
         for phonumb: =1+(j-1)*10 to j*10 do
           write(sm.nxtst[edstate,phonumb-1]:4);
           writein();
           writeln();
      end;
  wrtst;
  wrtword:
   curspos(phonum);
```

end;

The editor procedure allows the user access to all portions of the state machine to permit modification of the next states and outputs. The constants declarations it contains imbedded control characters which are specific to cursor movement from the Apple keyboard. The declarations are the characters for up, down, right, and left cursor movement.

```
Mover is passed characters which are keyboard inputs
        intended to move the cursor.
                                         It uses modulo
        arithmetic to calculate which phoneme the cursor has
                        This information is calculated
        been placed on.
        based upon the fact that there are 10 phonemes
         displayed on each line of the screen display. After
        calculating the new cursor position, the cursor is
        moved by calling curspos.
procedure mover(var ch:char);
begin
  case oh of
     down: begin
             phonum:=(phonum+10) mod (maxphons+1);
             curspos(phonum);
          end;
       up: begin
             phonum: = (phonum+30) mod (maxphons+1);
             curspos(phonum);
           end;
    right: begin
             phonum: = (phonum+1) mod (maxphons+1);
             curspos(phonum);
           end;
     left: begin
             phonum: =(phonum+maxphons) mod (maxphons+1);
             curspos(phonum);
           end;
  end;
end;
```

The statimp procedure permits modification of each next state while in the edit mode. Statimp determines which next state to alter based upon the cursor position. Statimpm is called by the main editor procedure when the character from the keyboard is a numeral. When the input is a numeral, there is an assumption that the next state pointed to by the cursor is to be changed. Once statimp is entered, the procedure performs a character to integer conversion in the repeat until loop until the user moves the cursor with a return, a right or left arrow, or a control O or L (Cntrl-O and cntrl-L move the cursor up and down respectively). Once entered the procedure reads the keyboard directly.

procedure statimp(var ch:char);

var
 tot:integer;
 quit:boolean;

begin
 quit:=false;
 tot:=0;
 repeat

{

 The character to integer conversion is accomplished by by taking the ordinal value of the input

character and subtracting the form it the ordinal walue of the character "0" which is decimal 48. The mod 256 operation prevents the input from exceeding the maximum state number and crashing the program.

tot:=(tot*10 + ord(ch)-48) mod 256; write(tot:4); curspos(phonum); read(keyboard,ch); case ch of up,down,right,left:quit:=true; end;

```
(
       Set the next state to the value obtained from the
       character to integer conversion.
sm.nxtstledstate,phonuml:=tot;
  until quit or eoln(keyboard);
If the input number was terminated by a return, then
       increment the phoneme number, i.e. move the cursor
      to the next higher numbered input phoneme. If the
      return was not used then we got an up, down right,
       or left cursor movement. Update the cursor position
       with mover.
  if not quit then phonum: =(phonum+1) mod (mamphons+1)
  else mover(ch);
  curspos(phonum);
end:
begin
                       (*editor*)
  promptline:=22;
  phonum: =0;
  pntscrn;
  quit:=false;
 repeat
    read(keyboard,ch);
    case ch of
      's', 'S': begin
Get an integer from the user to indicate which state
                               After getting the
       is to be viewed or edited.
       integer, update the screen.
promptat(promptline,'which state? ');
               readIn(edstate);
               edstate:=edstate mod 256;
               pntscrn;
             end;
      'q', 'Q': begin
```

```
Leave the editor and return to the procedure which
       called the editor.
                 quit: =true;
              end;
       'o','O': begin
(
       This function wants an integer. I have assigned
       word 0 to be "no output". The integer is used as an
        index into an array of characters to display the
        text of the output word. To display the output word
        "sero", enter a "1" here, "one" is displayed by
        entering "2", etc.
promptat(promptline, 'What is output word number? ');
                 readin(wordnum);
                 gotoxy(0,promptline);
                 crt(eraseol);
                 wordnum: =wordnum mod 41;
                 sm.wordledstatel:=wordnum;
                 wrtword;
               end;
       '+',';': begin
        Increment edstate and update the screen to show the
        new state. Edstate is the variable which determines
        the current state of the state machine while editing
        or executing.
  edstate:=(edstate +1) mod 256;
                 pntsern;
               ená;
```

'-': begin

```
Decrement edstate and update the screen.
                    edstate:=(edstate + 255) mod 256;
                    pntsorn;
                 end;
  up,down,right,
           left: mover(ch);
101,111,121,131,141,151,
'6','7','8','9' : begin
         There is a presumption that the next state is to be
         altered when the input is a numeral.
                    statimp(ch);
                 end;
      end; (*case*)
   until quit;
                              (*editor*)
end;
```

Statmach is the procedure which exercises the state machine. As mentioned previously, the variable edstate holds the current state of the machine. This program began life simply as an editor for the state machine. It became apparent that many benefits might be derived from combining the execution and editing functions.

The integer, laststat, holds the last state that the state machine was in. If laststat does not equal edstate, the screen is updated to display the current state. Otherwise, the screen is undisturbed. This is done because rewriting the screen is time consuming.

pracedure statmach;

VA.

oh: char;
running,goodnum,quit:boolean;
laststat,inphon:integer;

```
Procedure run is used to open a test file and set up
        conditions necessary to read input phonemes from
        that file. The name of the file is obtained from
        the keyboard. The procedure is repeated until a
        file is successfully opened or the user enters an
        empty line. A empty line is a return with no
        preceding characters.
procedure run;
 name:string;
 j:integer;
begin
  clear;
  repeat
     writeln('Which file?');
     readin(name);
     j:=length(name);
     (*$i-*)
     reset(inphons, name);
  until (j=0) or (ioresult=0);
ſ
Set the boclean variable, running, true. The
        variable, lines, is used to display the number of
        phoneme inputs read from the input file.
  running:=true;
  lines:=0;
end;
```

The getphon function probably contains the most obsure code in this program. While reviewing this procedure, keep firmly in mind the function of the procedure. Its function is to obtain an integer, result, to pass back to the procedure statmach. Result is an integer which is the number of the current input phoneme. The input phoneme may come from the keyboard or a file. The boolean variable, running, determines whether getphon goes to the keyboard or a file for the phoneme. Getphon is a boolean function. The value it returns indicates whether or not a good number has been obtained. A good number has not been obtained when the user enters anything other than digits. Running is set to false by reaching the end of the input file or detecting that a key has been pressed.

function getphon(var ch:char;var quit:boolean;var result:integer):boolean; var

Ĺ

Number is used internally by getphon to determine the final result of the function. Number simply indicates whether getphon has processed an integer or called another function such as the editor. While in getphon, if all keyboard entries are digits, number will remain true.

Savestat is used to remember the current state the machine has reached while executing. Savestat is used only when getphon calls the editor. Use of this variable permits the user to step around through the editor and make changes but upon exiting the editor, the machine returns to same state it was in on entry to the editor.

)

savestat:integer; number:boolean; first,second:integer; third:real;

```
begin
  result:=0;
  number: = true;
  if not running then
     begin
        prompt;
        repeat
          read(keyboard,ch);
           case ch of
        101,111,
        12','3',
        141,151,
        161,171,
        '8','9':begin
(
        Perform character to integer conversion
}
                  result:=result*10+ord(ch)-48;
                  if result(1 then result:=1;
                  if result)40 then result:=40;
                  gotoxy(37,23);
                  write(result:2);
               end;
        'e', 'E':begin
         Call the editor
                  savestat: =edstate;
                  xmode:=false;
                  editor;
                  zmode:=true;
                  if edstate()savestat then
                     begin
                        edstate:=savestat;
                        pntscrn;
                     end;
                  promptat(0,'Execution mode');
                  number:=false;
               end;
        'q','Q':begin
```

A STATE OF THE STA

```
Quit the getphon procedure and set quit true. This
      will cause an exit from the statmach procedure.
      Leaving statmach produces a return to the main loop
      of the program.
 number:=false;
             quit:=true;
             running: = false;
      'r', 'R':begin
Call run and open a phoneme input file.
3
             run;
           end:
                       (*case*)
        end:
      until coln(keyboard) or not number;
      if number and (result)0) then
        begin
         result:=result-1;
          getphon:=true;
        end
      else getphon:=false;
    end
                     (* running is true *)
  else
    begin
(
       If we haven't reached the end of the file and no key
      has been pressed then read a phoneme from the input
      file and pass it back to statmach. If a key has
      been pressed or we've reached the file end, then set
      running false and close the input file. With
      running false, the next entry to getphon will have
      getphon expectig input from the keyboard.
 if not eof(inphons) and not keypress then
        begin
```

readin(inphons, first, second, third);

```
result:=first;
    getphon:=true;
    lines:=lines+1;
    gotoxy(33,23);
    write(lines);
    gotoxy(37,23);
    write(first:2);
    end
    else
     begin
        running:=false;
        close(inphons,lock);
    end;
end;
```

```
Statmach is the main procedure which exercises the
         state machine. It remains in a loop of getting an
         input phoneme and and then interrogating the next
         state table and then returning for a new input
         phoneme. While in the procedure, getphon, it is
         possible to enter the editor and modify the next
         state table.
begin
                               (* statmach *)
  edstate:=0;
  laststat:=0;
  quit:=false;
  running:=false;
  clear;
  if visible then pntscrn;
  repeat
      goodnum:=getphon(ch,quit,inphon);
      if goodnum then
         begin
            edstate:=sm.nxtst[edstate,inphon];
            if visible and (laststat()edstate) then
               begin
                  pntscrn;
                  laststat:=edstate;
                end:
             if not visible then
               begin
                  promptat(18, 'output word is ');
                  write('I', sm.wtext[sm.word[edstate]],']');
                end;
         end;
  until quit;
```

(*statmach*)

end;

```
The vocab procedure permits defining the text of
         each word in the vocabulary of the state machine.
procedure vocab;
149
  wordnum, i, 1: integer;
  name:string;
begin
  repeat
     clear;
     promptat(0,'Vocabulary text entry. Neg # to exit.');
     promptat(1,'Which word number?');
     writeln();
     readin(wordnum);
     if (wordnum)=0) and (wordnum(=40) then
        begin
           promptat(1,'Word ');
           write(wordnum,' is ','[',sm.wtext[wordnum],']');
           promptat(2, 'Enter text. Max is 10 characters');
           writeln();
           readin(name);
           1:=length(name);
           if 1>0 then
             begin
                for i:=1 to 1 do
                   sm.wtext[wordnum,i]:=name[i];
                for i:=1+1 to 10 do
                   sm.wtext[wordnum,i]:=' ';
             end;
        end:
  until wordnum(0;
end;
```

```
The print procedure permits printing the next state
         information for an arbitrary number of states. The
          information is printed in the format used for the
         screen display.
                           Three screens of information are
         printed on each page.
************************************
3
procedure print;
   j,k,phonumb:integer;
   counter, state, start, endst: integer;
   answer: char;
  printer: text;
begin
   { * i = * }
   rewrite(printer, 'printer: ');
   (*i+*)
   if ioresult=0 then
      begin
        writeln;
        write('Enter beginning state to print ');
         readIn(start);
        write('Enter end state to print ');
        readin(endst);
         counter: =-1;
        write(chr(12));
         for state:=start to endst do
            begin
               counter:=(counter + 1) mod 3;
               if counter=0 then
                  begin
                    for k:=1 to 7 do
                       write(printer,chr(13));
                  end;
               for j:=1 to 4 do
                  begin
                    for phonumb: =1+(j-1)*10 to j*10 do
                       write(printer,phonumb:4);
                       writeln(printer);
                     for phonumb:=1+(j-1)*10 to j*10 do
                       write(printer, sm.natst[state, phonumb=13:4);
                       writeIn(printer);
                       writeln(printer);
                  end;
```

```
writeln(printer, 'State: ', state);
                 writeIn(printer, 'Output word is I',
                          sm.wtext[sm.word[edstate]],']');
                  for k:=1 to 4 do
                     write(printer,chr(13));
                  if counter=2 then write(printer,chr(12));
            end;
         if keypress then
           begin
               writeln;
               write('Quit printing? y/n');
               readin(answer);
               if (answer='y') or (answer='Y') then
                  begin
                     writeln(chr(12));
                     exit(print);
                  end;
            end;
      end;
end;
```

```
The main procedure is a keyboard input routine
          followed by a case statement. The user may edit or
          execute from the main loop.
}
begin
  init;
   repeat
      screen;
      ch:=getchar(['q','G','p','P','x','X','e','E','v','V','w','W']);
      case ch of
         'e', 'E':begin
                    mmode:=false;
                    editori
                 end;
         'p', 'P':print;
         'q','Q':begin
                    quit:=true;
                 end;
         'x','X':begin
                    visible:=false;
                    zmode:=true;
                    phonum: =0;
                    statmach;
                 end;
         'v','V':begin
                    visible:=true;
                    zmode:=true;
                    phonum: =0;
                    statmach;
                 end;
         'w', 'V': begin
                    vocab;
                 end;
      end;
              (#case*)
   until quit;
   Savesm;
   clear;
```

end.

The following intrinsic unit is a separately compiled unit which must be placed in the system library for the main program to use. Alternatively, the procedures may be extracted from their context in this intrinsic unit and compiled as in line procedures in the main program.

The following procedures were extracted from a program titled, DISKIO, on the APPLE3: diskette supplied with the Apple Pascal system. The procedures are useful in that they provide terminal independent screen output as well as "bullet proof" input by the procedure getchar;

```
($s+)
unit io; intrinsic code 25 data 26;

interface
type
    crtcommand = (eraseos,eraseol,up,down,right,left,leadin);
    setofchar = set of char;
var
    f: file;
    crtinfo: packed array[crtcommand]of char;
    prefixed: array [crtcommand] of boolean;

    procedure beep;
    procedure crt(c: crtcommand);
    procedure promptat(y: integer; s: string);
    procedure clear;
    function getchar(okset: setofchar): char;

implementation
    var
    buffer: packed array[0..511] of char;
```

i,byte: integer;

6-1

```
Getchar returns a single character. The programmer
         specifies an acceptable set of inputs when the
         procedure is called. If a user attempts to type a
         character not specified by the programmer, getchar
         beeps and waits for a proper input.
function getchar;
(* get a character, beep if not in okset, echo only if printing *)
var ch: char;
   good: boolean;
begin
 repeat
   read(keyboard,ch);
   if eoln(keyboard) then ch:=chr(13);
   good: = ch in okset;
   if not good then beep
      else if ch in [' '..'}'] then write(ch);
  until good;
  getchar: =ch;
end;
```

The code in this main procedure of the unit is run automatically by the Pascal system before the main program is loaded. This code reads the file, SYSTEM MISCINFO to obtain the information necessary to control the terminal. ************************* begin reset(f,'*system.miscinfo'); i:=blockread(f,buffer,1); close(f); byte:=ord(buffer[72]); (* prefix information byte *) crtinfolleadin]:=buffer[62]; prefixed[leadin]:=false; crtinfoleraseosl:=buffer[64]; prefixed[eraseos]:=odd(byte div 8); crtinfoleraseoll:=buffer[65]; prefixed[eraseol]:=odd(byte div 4); crtinfo[right]:=buffer[66]; prefixed[right]:=odd(byte div 2); crtinfo[up]:=buffer[67]; prefixed[up]:=odd(byte); crtinfolleft]:=buffer[68]; prefixed[left]:=odd(byte div 32); crtinfoldown]:=chr(10); prefixed[down]:=false; end.

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<u>Vita</u>

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ABSTRACT (Continue on reverse side it necessary and identify by block number) Thesis attempts continuous speech recognition by parsing phoneme strings to words. The phoneme strings are obtained from the Air Force Institute of Techonology acoustic analyzer. Vocabulary is limited to the 10 digits. Recognition is accomplished by a software simulation of a finite state markinexautomata. State tables are read in as a datasfile and manipulated without recompiling.	

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